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DEVELOPMENT OF A SPEECH-TO-NOISE RATIO MEASUREMENT UTILIZING DIGITAL TECHNIQUES

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FOREWORD

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I. INTRODUCTION

General

Communications engineers and research scientists have conducted numerous tests and experiments in order to quantify the effects of noise on speech communications. The state of knowledge is such that, on the basis of physical measures of the noise and speech spectra, one can, under certain conditions, calculate approximately how much sound control must be applied and what the characteristics of the communication system must be in order to achieve satisfactory communications in noisy situations. However, the methods developed to date takes into account only certain physical attributes of the noise and speech and can handle, with real validity, only a limited number of conditions. Each operational situation we wish to evaluate with respect to speech communication will bear analysis of many psychological and physiological as well as acoustical factors.

Background

The Apollo voice communications system has been undergoing extensive testing to insure reliability and compatibility between the Extra Vehicular Activity (EVA), manned spacecraft and the Manned Space Flight Network (MSFN) ground stations. This testing is performed in the Electronic Systems Compatibility Laboratory (ESCL) of the Information Systems Division. A monosyllabic word intelligibility (WI) measurement technique is being used to measure and predict the actual performance of the voice communication links (reference 1 of Appendix A).

The objective, to verify the performance and compatibility of the voice communication links, is accomplished by measuring the signal power of a constant 1 KHz tone and noise power at the audio outputs of the ground station and flight equipment, and recording the voice outputs of the system for different modes of operation. The voice output tapes are scored by trained subjects to obtain quantitatively the percentage of word intelligibility. These intelligibility scores are plotted versus the measured signal-to-noise ratio (S/N) on a per link basis and evaluations are made of the predicted performance of these links with this data. On numerous tests it was found that the data relating signal-to-noise ratios to percent word intelligibility did not follow definite trends and there was a scattering of data points over the entire range of interest. The main reasons for the poor correlation between S/N and W/I are as follows:

a. The present S/N measurements, made with the 1 KHz does not account for changes in the input speech level from the audio source tapes. The resulting WI scores are a function of the input level.

b. The present measurements do not account for the presence of suit noise on the source tape. The resulting WI scores are a function of the inclusion of this noise.

c. The present measurements, do not account for voice distortion due to non-linear speech processing such as peak or center clipping while the resulting WI scores are a function of the voice distortion.

Due to the problems associated with relating signal-to-noise ratios with percent WI it was intuitively felt that a measurement relating acoustical factors of the voice channels to percent word intelligibility would produce more realistic results.

It was decided that a measurement of the ratio of speech power to noise power was required to accurately correlate the voice channels acoustical characteristics. A measurement of this type is also required to give a quantitative performance evaluation of voice communication links on future manned missions.

A speech-to-noise ratio (SPNR) measurement is very difficult to make using ordinary amplitude detection devices, such as RMS meters because of the slow response and narrow dynamic range. More accurate and repeatable time varying measurements may be made by digitizing the speech and noise and performing the necessary calculations using a digital computer. The digital method would also facilitate the measurement and evaluation of numerous amounts of tapes now stored in the Audio Facility Library and voice tapes from future tests while conserving manhours.

II. SUMMARY

As a result of the inability to obtain accurate correlation between signal-to-noise ratio and percent word intelligibility, a task was initiated by the Information Systems Division (ISD) to develop a speech-to-noise ratio measurement method utilizing digital techniques. The developmental phase consisted of designing a measurement method, determining its validity and configuring necessary system software and hardware to accomplish the measurement. This report describes the task accomplishments in the progression that they were undertaken.

In order to efficiently utilize the time and to ensure a complete and accurate study, the general requirements for the desired measurement were established. These requirements outlined desirable characteristics for the systems hardware and software.

A study of existing documents on speech characteristics and the effects of noise on speech intelligibility was made. An analysis of the unique features of speech and noise that was applicable to the development of speech-to-noise ratio measurements was also made. It was determined that the uniqueness of vowel sounds being the longest and strongest and characterized by continuous rather regular wave trains was the principal feature that the measurement method could be built around. The characteristics of vowel sounds was the key to the design of a detection scheme to sort speech power from speech plus noise power.

A logical scheme was designed that produced a speech-to-noise ratio as a function of time. The measurement scheme consisted of comparing the mean square values of at least three consecutive 20 msec intervals of digitized data for an agreement within 1 dB. The average value of the consecutive intervals that agreed were calculated and placed in storage until 1 second of raw data went through the comparing process. The values in storage were then rearranged into algebraic order and the lowest or first value defined as a noise value. This noise value was then used as a

reference to compare the other values with. If a value compared within 1 dB it was defined as noise. If the value was greater than 3 dB it was defined as speech plus noise. All the noise values were averaged and all the speech plus noise values were averaged and a speech-to-noise ratio was calculated. The process was designed for continuous repetition of the above steps, to obtain speech-to-noise ratios for every second of data taken.

Prior to implementing the digital hardware and software, an analog system was configured and used to simulate the digital system in order to verify the proposed measurement techniques. Use of the analog system permitted personnel to gain experience with the techniques and determine what changes would have to be made. The verification procedure used followed the above described scheme for the detection of speech from speech plus noise. By using high speed recording and slow speed playback, an effective reduction of 64 times was achieved. This data was rectified with a mean square meter, measured with an integrating digital voltmeter and printed out on paper tape. The sampling rate of the digital voltmeter and printer was adjusted to simulate measured values of 20 msec real-time. The calculated values of speech-to-noise ratios were in close agreement with expected values and the detection scheme was able to discriminate speech power from tapes containing very high noise content.

After the measurement approach had been verified by the analog simulation, the digital hardware was selected and the equipment configuration determined. The primary consideration for the equipment chosen was immediate availability in the Data Systems Development Lab of ISD while still meeting established criteria. The analog to digital conversion system was configured with a Magnecord model 1048 audio recorder and a Systems Engineering Labs (SEL) model 600 data acquisition system. It was decided to use an IBM 360 model 44 computer to run the main measurement and calculation program; however, the IBM format was not compatible with the SEL 600 format. Therefore, the Data Machines, Inc. (DMI) model 620 computer was selected as an input-output (I/O) system to reformat digital tapes.

Once the hardware configuration was determined, the software requirements were implemented into usable programs. The reformat program was written for the DMI 620 in machine language and the IBM 360/44 program was written using FORTRAN IV. All the preliminary schemes designed in the early stages of the development that were validated with the analog system were incorporated into the IBM system program.

Using the above described hardware and software configurations, speech-to-noise ratio tests were run under varying conditions, to disclose any hardware or software problems and to determine the performance of the system parameters. The results of the tests that were run indicated that the hardware constraints could be tolerated in the development of the measurement method and that the IBM program, required further refinements. The areas of refinement are generally in the level detection scheme to accurately discriminate between speech and speech plus noise under any condition. Also, additional statistical techniques must be incorporated in the program to more accurately identify the speech-to-noise ratio over an optimum defined period of time.

It was concluded from the results of the study that the general requirements and objectives for a speech-to-noise ratio measurement utilizing digital techniques can be attained. Results to data also indicate that a more accurate correlation to percent word intelligibility can be made using the speech-to-noise ratio measurement method than the presently used signal-to-noise measurement approach.

Recommendations were made to initiate a Phase II development task in order to further refine the speech-to-noise ratio measurement method to operational status. The digital system configuration that was used in the inital development of the measurement method should be used for the advanced phase. It was also recommended to develop a digital statistical correlation system constructed in conjunction with the DMI 620 computer. The requirements of this proposed system are to complement the present measurement approach and to develop additional digital voice analysis techniques.

III. SPEECH-TO-NOISE RATIO (SPNR) DEVELOPMENT

The development of a method for measuring a SPNR utilizing digital techniques was initiated by establishing general requirements for the entire system. A study was made of applicable information available on speech and information theory and a preliminary scheme was designed to separate speech plus noise and noise segments and calculate a SPNR.

General Requirements for Measurement Techniques

The general requirements that were considered desirable for the development of a digital system that would measure SPNR as a function of time are as follows:

a. The discrimination between speech and noise should be made using the system software in order to avoid the development of any voice detection equipment and interfaces associated with this equipment.

b. The noise measurement associated with the SPNR calculation should be obtained from in between syllables and in between words since this is the noise that will be competing with or masking the speech.

c. Minimize or eliminate the human decisions or observations in order to obtain a repeatable measurement.

d. A continuous loop technique should be determined. Once the system is initiated and the analog data is fed into the system, it will continue to run and output SPNR's as a function of time until the input is stopped.

e. The systems software should be compatible with any format of voice input.

f. Output a SPNR in a short enough time interval so that the voice channel data could be evaluated during launch and staging phases of manned missions.

Characteristics of Speech

In order to develop a valid SPNR measurement, a study of the acoustical characteristics of speech must first be made and the unique features associated with speech be applied to the measurement.

Speech Spectra:

One of the primary methods of determining the acoustical properties of speech is by a frequency-amplitude analysis of continuous speech averaged over a long period of time. Investigations have determined that intervals of 30, 60, and 90 seconds to be sufficiently long for this purpose. Major frequency-amplitude speech spectra measurements have been made by Crandall and MacKenzie, Dunn and White, Rudmose et al, and Benson and Hirsh (references 2, 3, 4, and 5 respectively of Appendix A). In these studies, various speech materials were uttered at normal voice levels in a free-field condition. Typical results of their studies are shown in Figure 1. Although the speech spectra obtained by Dunn and White, as shown in Figure 1, are commonly accepted and used as typical, the bandwidth filters used "averaged out" the true variations per cycle in the speech spectrum. The true speech spectrum actually fluctuates widely as a function of frequency and does not look like the smooth curves shown in Figure 1.

Distribution of Speech Energy:

Dunn and White, in addition to obtaining the long time spectrum, made measurements of the distribution of sound pressure per cycle of continuous speech for 1/8-second intervals. The samples were taken over a 75-second period of continuous speech. Continuation beyond 75 seconds did not increase the spread of the distribution. 1/8-second was chosen as representative of the length of a syllable.

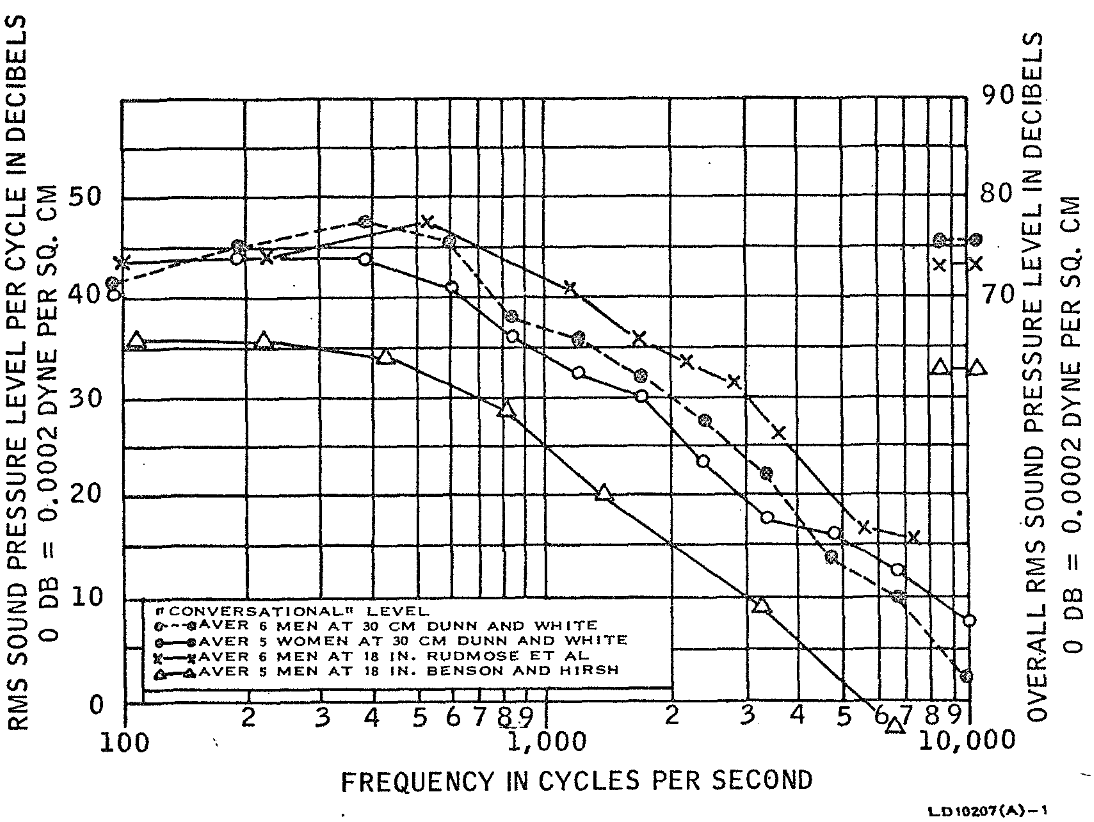

Figure 1 Average Speech Spectra Obtained by Dunn and White, Rudmose et al and Benson and Hirsh. (The Speakers for each study attempted to use a so-called normal conversational effort. The overall sound pressure levels are given on the ordinate on the right-hand side.)

The use of 1/4-second intervals gave essentially the same results. Figures 2 and 3 show the peak pressures and RMS pressures respectively in 1/8-second intervals. It is apparent that the listener must detect sound throughout the dynamic range of about 42 dB at each frequency (the average difference between the lowest and highest peak contours) in order to hear all of the components of conversational speech.

Average Power in Speech Sounds:

Figures 1, 2, and 3 give a statistical evaluation of the frequency-intensity characteristics of conversational speech, averaging overall speech sounds and natural periods of silence. Many of the same or similar measures have been applied directly to speech sounds individually and are of interest to the development of a SPNR.

Table 1, taken from Sacia and Beck (reference 6 of Appendix A), contains the following determinations for the primary speech sounds:

a. Phonetic power, the maximum value of the mean power during the duration of the individual speech sounds listed.

b. Peak power achieved during the speaking of the individual sounds.

c. The average and maximum values for the speakers.

For these measurements, 16 people were used and the sounds were uttered in various syllabic context. Considering these and other data, Fletcher (reference 7 of Appendix A) concludes that for one speaker there is a range of approximately 28 dB between the weakest and the most powerful speech sound in terms of mean power. Table 1 reveals about 12 dB peak factor (peak-to-mean) that gives us an over-all range of about 40 dB from the weakest (th in thin) to the strongest (o' in talk) sound. This compares well with the 42 dB range arrived at from consideration of the measurements made by Dunn and White using continuous speech.

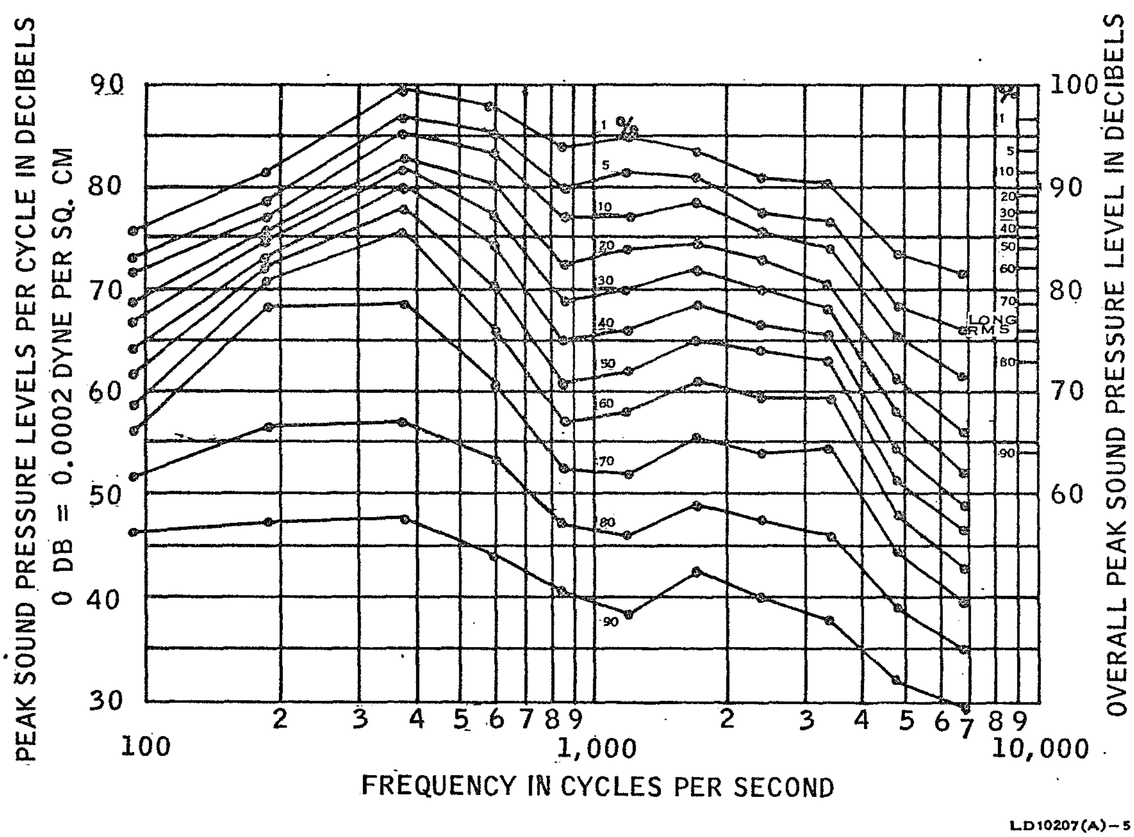

Figure 2 Peak Pressure in 1/8 sec. Intervals for Conversational Speech at a Distance of 30 cm from the Speakers 'Lips (Average of Six Men). From Dunn and White.

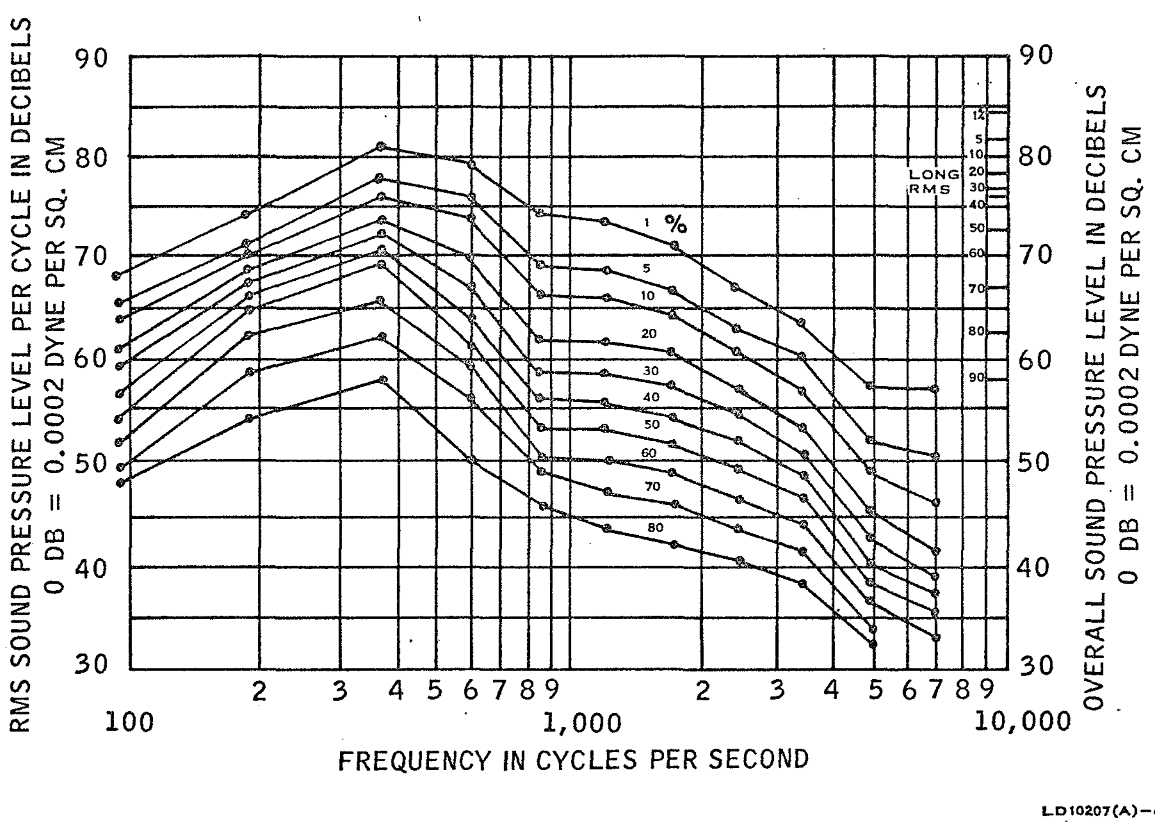

Figure 3 RMS Pressures in 1/8 sec. Intervals for Conversational Speech at a Distance of 30 cm from the Speakers' Lips (Average of Six Men). From Dunn and White.

TABLE 1

POWER IN MICROWATTS IN THE FUNDAMENTAL SPEECH SOUNDS

PHONETIC SOUND	KEY WORD	PHONETIC POWER		PEAK POWER	
		AVERAGE	MAXIMUM	AVERAGE	MAXIMUM
ū	tool	23	60	235	700
u	took	26	100	470	890
ō	tone	25	80	435	1300
o'	talk	45	120	615	1500
o	ton	24	110	450	1700
a	top	41	120	700	1600
a'	tap	25	90	650	1800
e	ten	22	90	500	1700
ā	tape	23	60	525	1700
i	tip	20	50	350	1300
ē	team	20	80	310	1500
m	me	1.8	17	110	200
n	no	2.1	18	47	70
ng	ring	0.3	3.6	97	170
l	let	0.3	9.6	130	230
r	err	16	30	200	600
v	vat	0.03	2.4	25	30
f	for	0.08	3.6	3	4
z	sip	0.7	7.2	30	40
s	sit	0.9	8.7	30	55
th	thin	.....	.....	1	1
$\underline{\text{th}}$	that	.....	.....	9	10
zh	asure	.....	.....	40	55
sh	shot	1.8	6.0	110	130
b	bat	.....	.....	7	7
p	pat	.....	.....	6	7
d	dot	0.08	2.9	4	7
t	tap	0.1	6.0	16	19
j	jot	0.5	3.6	24	36
ch	chat	1.4	19	52	60
g	get	.....	.....	8	9
k	kit	.3	4.8	6	9

It is apparent from Table 1 that the major portion of the wide dynamic range of conversational speech is due to the general low intensity of the consonants in comparison to the vowel sounds. The vowel sounds contribute over 90% of the total power spectrum. In general, vowels are characterized by continuous, rather regular wave trains formed in the throat and passing through open passages (reference 8 of Appendix A).

Amplitude-Time Analysis:

In an attempt to take advantage of the fact that vowel sounds are characterized by continuous rather regular wave trains, amplitude time plots were run and analyzed. Figure 4 is a typical waveform of a male speaker saying the word "top". The lower trace represents the audio input and the upper trace is the audio output. This particular waveform exhibits peak clipping of the original input waveform. The center portion of the word contains the vowel sound /a/ or phoneme "o". The duration of the vowel sound is approximately 120 milliseconds. The stop consonants /t/ at the beginning of the word and /p/ at the end of the word are seen to be of much shorter duration and very irregular in waveform. This is the typical difference between vowels and consonants. The vowels are not only the strongest sounds but are also the longest. Table 2 presents statistical data taken from 104 records of vowel sounds (reference 9 of Appendix A).

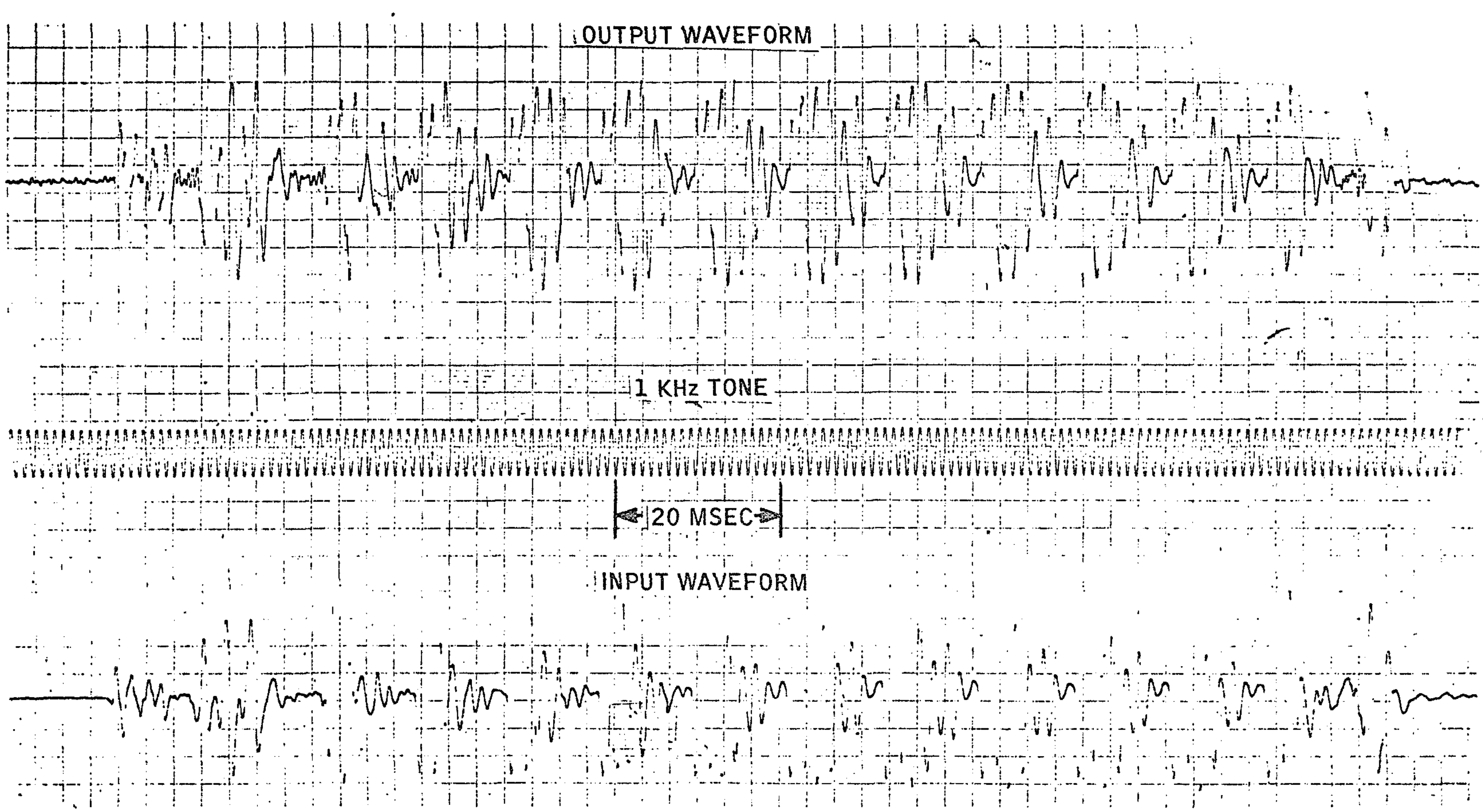

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Figure 4 Amplitude-Time Analysis, Speech Waveform Subject saying the word "Top".

TABLE 2

STATISTICAL DATA FROM 104
RECORDS OF VOWEL SOUNDS

	SOUND		TOTAL DURATION (SEC.)	START (SEC.)	MIDDLE (SEC.)	DECAY (SEC.)
I	ū	(pool)	0.351	0.061	0.164	0.126
II	u	(put)	0.349	0.057	0.115	0.077
III	ō	(tone)	0.325	0.053	0.139	0.133
IV	o'	(talk)	0.290	0.034	0.191	0.065
V	o	(ton)	0.280	0.046	0.179	0.061
VI	a	(father)	0.306	0.029	0.199	0.078
VII	ar	(part)	0.345	-----	-----	-----
VIII	a'	(tap)	0.294	0.038	0.180	0.076
IX	e	(ten)	0.219	0.034	0.119	0.066
X	er	(pert)	0.331	-----	-----	-----
XI	ā	(tape)	0.305	0.042	0.172	0.091
XII	i	(tip)	0.211	0.036	0.126	0.049
XIII	ē	(team)	0.341	0.036	0.189	0.116

Characteristics of Noise

The effects of noise on speech reception is that of producing frequency distortion depending on the noise spectrum, that is, certain frequency components of the speech signals are effectively degraded or their audibility supressed because of masking.

The noise bandwidth of most Apollo voice channels is approximately 4 KHz. Recently, some spectral analysis work has been performed on the Apollo Communication links (references 10 and 11 of Appendix A). The results indicate that there are basically two types of noise spectrums in the voice channel bandwidth. There is essentially a flat band spectrum or "white noise", and an exponential-shaped band spectrum or, "FM noise". The exponential-shaped noise spectrum increases in amplitude from the low frequency edge of the band to the high frequency end. Suit noise, caused by the passage of air into the space suit helmet, occurs generally between 1 to 4 KHz. The suit noise peaks the flat or FM spectrum in this narrow band.

For the development of a SPNR, working in the time domain would appear to contribute an indeterminable amount of error if it was assumed that the noise spectrum is flat in the frequency domain. However, by measuring the SPNR on a per link basis and correlating the measurement to the tested WI scores, it is intuitively felt that the shaped noise spectrums will be accounted for much the same way as the speech spectrum distortion.

Designed Speech-to-Noise Ratio Method

The philosophy used in designing a test method of this kind is to obtain a relationship between the two assumed variables (speech and noise power). The relationship to be obtained will be the ratio of the speech power to the noise power. An exact physical description of each variable such as measurement of the power in microwatts is not necessary. Also, insuring repeatability and consistency in the method of determining the relationship insures a valid method. Being cognizant of the characteristics of speech and noise and the usable statistical approaches and techniques as outlined in Appendix B, a unique and well grounded scheme was designed to detect and separate the speech plus noise and noise components and calculate an SPNR.

Detection Scheme:

The design of the detection scheme was the first problem to surmount, and was based on the previously explained relationship found between the vowel sounds and the total speech spectra. Remember, the vowel sounds are the longest and strongest, and are characterized by continuous, rather regular wave trains and are of an intensity to represent the speech power spectrum.

Assume that the average power of the vowel plus noise waveform will not deviate more than 1 dB throughout the duration of the sound. It can also be assumed that the in-between syllable and in-between word noise (average power) will not deviate more than 1 dB for approximately the same time interval as a vowel sound. It can be seen from Table 2 that the minimum middle duration of a vowel sound is approximately 100 msec and the maximum is approximately 200 msec.

Consider, that in order to insure that the sample was either a noise segment or a vowel plus noise segment of conversational speech plus noise, the average power in consecutive time intervals must be compared to 1 dB within a duration of 100 msec. The number of intervals chosen was 3 with a time duration of each interval of 20 msec. The duration of a minimum of 3 consecutive time intervals would fall safely within the 100 msec vowel sounds.

If 2 consecutive time intervals were chosen and the average power of the two-compared to each other, an agreement may be made due to coincidence. However, the probability of 3 consecutive 20 msec intervals agreeing due to coincidence was less likely. The interval was chosen to be 20 msec for the following reasons: (1). the interval had to be small enough to obtain at least 3 intervals during a vowel and (2). yet long enough to obtain the required digitized samples in the interval to adequately define the average power (e.g., 400 samples at 20 K samples/sec.).

Three or more consecutive intervals were averaged and placed in storage until 50 consecutive 20 msec intervals had been compared or for a total time duration of 1 second. At this time there was average noise power values and average vowel plus noise (speech plus noise) power values in storage. The consonants dropped out because their duration was less than 60 msec and it is unlikely that 3 consecutive average power agreements will be obtained during this occurrence.

Sorting Scheme:

In order to sort the speech plus noise from the noise power in the 1 second interval, a level detection technique was employed where it was assumed that the smallest value in storage was a noise value. The logic is that the noise values may possibly be increased through data processing but not decreased if the computations are valid. It was also assumed that noise exists in the 1 second interval. A comparison was made between all values in storage with the noise reference value. If a value compared with the noise reference within 1 dB it was sorted into a noise section. If a value was 3 dB or greater than the noise reference it was placed into a speech plus noise section. The values between 1 dB and 3 dB are ignored.

Final Calculations:

The mean (average) was calculated for all the values in the speech plus noise and noise sections. These average values were used in the calculation of the speech-to-noise ratio for the 1 second duration. The calculation of SPNR is,

$$\text{SPNR} = 10 \log \frac{(\text{Speech} + \text{Noise}) - \text{Noise}}{\text{Noise}}$$

The minimum SPNR that can be calculated is 0 dB and occurs when the average speech plus noise power is 2 times the average noise power. equivalent to [illegible]

The average noise power can be subtracted from the average speech plus noise power if we assume that speech and noise are random independent statistical variables.

Analog Verification

The previous devised detection scheme was verified by using an analog system that would simulate a digital system. Analog verification of the designed method was required, before implementing the digital hardware and software to gain experience with the method and to determine if any further changes would be necessary. A block diagram of the equipment configuration is shown in Figure 5. The theory of operation of the system is as follows:

An Apollo communications system test data tape that was recorded at 7-1/2 inches per second (ips) was played back at 3-3/4 ips and recorded on a wide-band tape recorder at 120 ips and played back at 3-3/4 ips. From end-to-end, the original analog waveform was slowed down or speed reduced 64 times. This time-reduced waveform was measured by a True Mean Square Meter. The output of the meter represents an analog squared or rectified version of the input signal by employing a square law detection ($i \alpha E^2$) circuit.

The mean square waveform was then fed into an integrating digital voltmeter. The sampling rate was adjusted to average the mean square signal over 1.28 seconds which represents 20 msec of real-time data (64 times). The digital voltmeter was coupled to a paper tape printer. The tape output is representative of the series of average values of 20 msec real-time intervals. Since the average mean square values are proportional to the voltage^2 (reference 12 of Appendix A), the final values are also proportional to the average power over 20 msecs. These values were then compared manually using the previously discussed detection scheme, sorting scheme and calculations to obtain a speech-to-noise power ratio for each 50 values or for a real-time duration of 1 second.

The speech-to-noise ratios obtained were in close agreement with the measured 1 KHz tone signal-to-noise ratio of the same tape. This testbed was also used to evaluate test data tapes that have high noise power, to validate the resolution of the detection scheme. Results indicate that speech-to-noise ratios of 0 dB can be achieved.

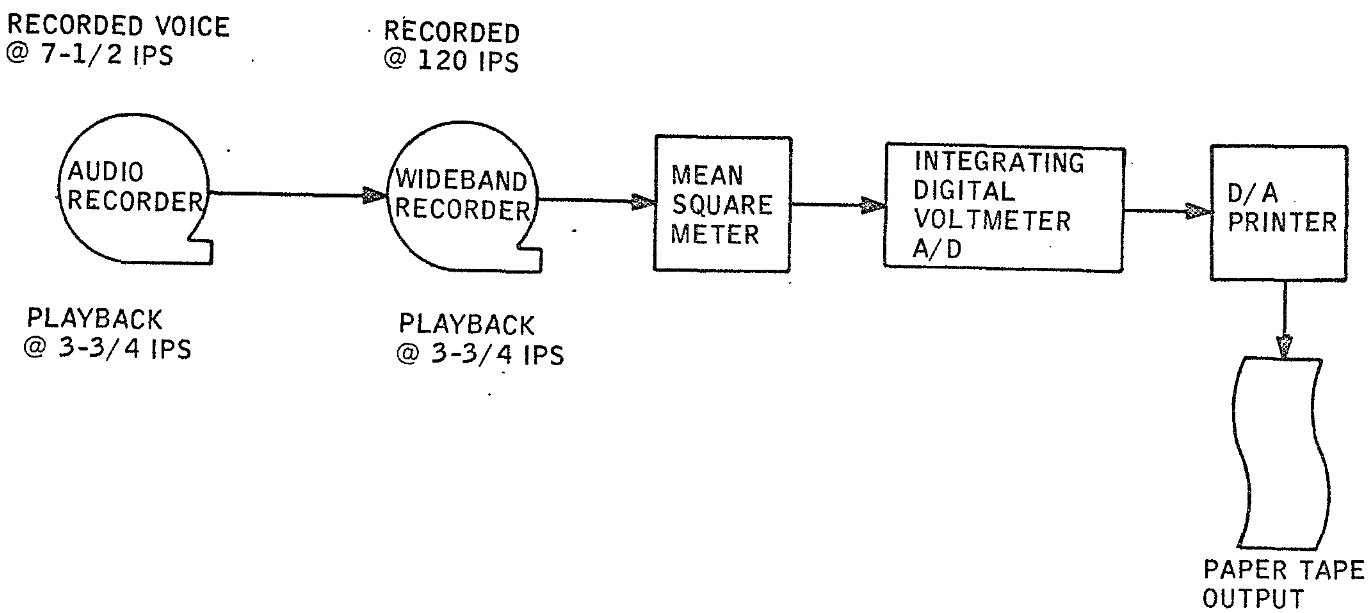

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Figure 5 Analog Verification Equipment Configuration

IV. SPEECH-TO-NOISE RATIO DIGITAL IMPLEMENTATION

To implement and analyze the scheme that was chosen, hardware and software requirements were established and configured and verification tests were run.

Hardware Selection and Configuration

An evaluation of required hardware to implement a measurement technique was made of the available equipment in the Data Systems Development Laboratory.

The requirements for an analog to digital conversion system (A/D) is based on two parameters: sampling rate and quantization (see Appendix B). The minimum sampling rate requirement based on the Nyquist frequency, is 8,000 samples/sec with a channel bandwidth of 4 KHz.

$$\text{Let } f_c = 4 \text{ KHz} = \frac{1}{2h}$$

$$\text{Sampling rate} = \frac{1}{h} = 8{,}000 \text{ samples/sec} \tag{10}$$

It is generally acceptable to digitize at 1-1/2 to 2 times the minimum rate to insure aliasing are eliminated. Therefore, the choice of an A/D converter was based on the capability of sampling at least 12,000 samples/sec.

The requirements for the dynamic range of the measurements was chosen to be at least 60 dB. The full range signal or quantization level must therefore be:

$$\text{RMS Signal-To-Noise Ratio} = 60 \text{ dB}$$

$$\begin{aligned}\text{Full Range Signal} &= 0.29 \times \text{RMS Signal-To-Noise Ratio}\\ &= 290\end{aligned} \tag{11}$$

A full range signal of 256 based on a binary equivalent of 2^n (where $n = 8$) would be satisfactory. The A/D converter should therefore have at least 8 bits plus sign. The polarity may be indicated by an extra bit or by using complementary 1's and 0's. The converter input must also be compatible with the analog playback electronics (e.g., output level and impedance). A block diagram of the digital test configuration is shown in Figure 6. A Magnecord model 1048 audio tape recorder was selected as the analog recorder because of its portability and immediate availability. The maximum output level of the recorder is 10 volts peak-to-peak into a 600 ohm load and its output impedance is a balanced 600 ohms.

The A/D conversion system selected to implement the SPNR development was the Systems Engineering Labs (SEL) model 600 data acquisition system. The SEL 600 is compatible with the Magnacord tape recorder. It has a maximum sampling rate of 15,120 samples/sec and digitizes with 11-data bits plus sign bit. The digital data output is written on magnetic tape. The immediate availability plus meeting all requirements made its selection practical.

It was decided to use the IBM 360/44, located in the Data Systems Development Laboratory, to calculate the SPNR. This system is best suited to a measurement development program of this type. The various subroutines required are readily able to be defined with the use of FORTRAN IV language. An efficient time saving program was not as important during the development as proving a workable technique. The system contains magnetic tape inputs and a line printer output.

Since the IBM 360/44 was selected to run the main program, an input-output (I/O) system was required to reformat the SEL 600 tapes to be compatible with the IBM 360/44. The Data Machines, Inc. (DMI) model 620 computer was selected and a machine language program was written to read the SEL 600 tape and write a tape in a compatible format for the 360/44.

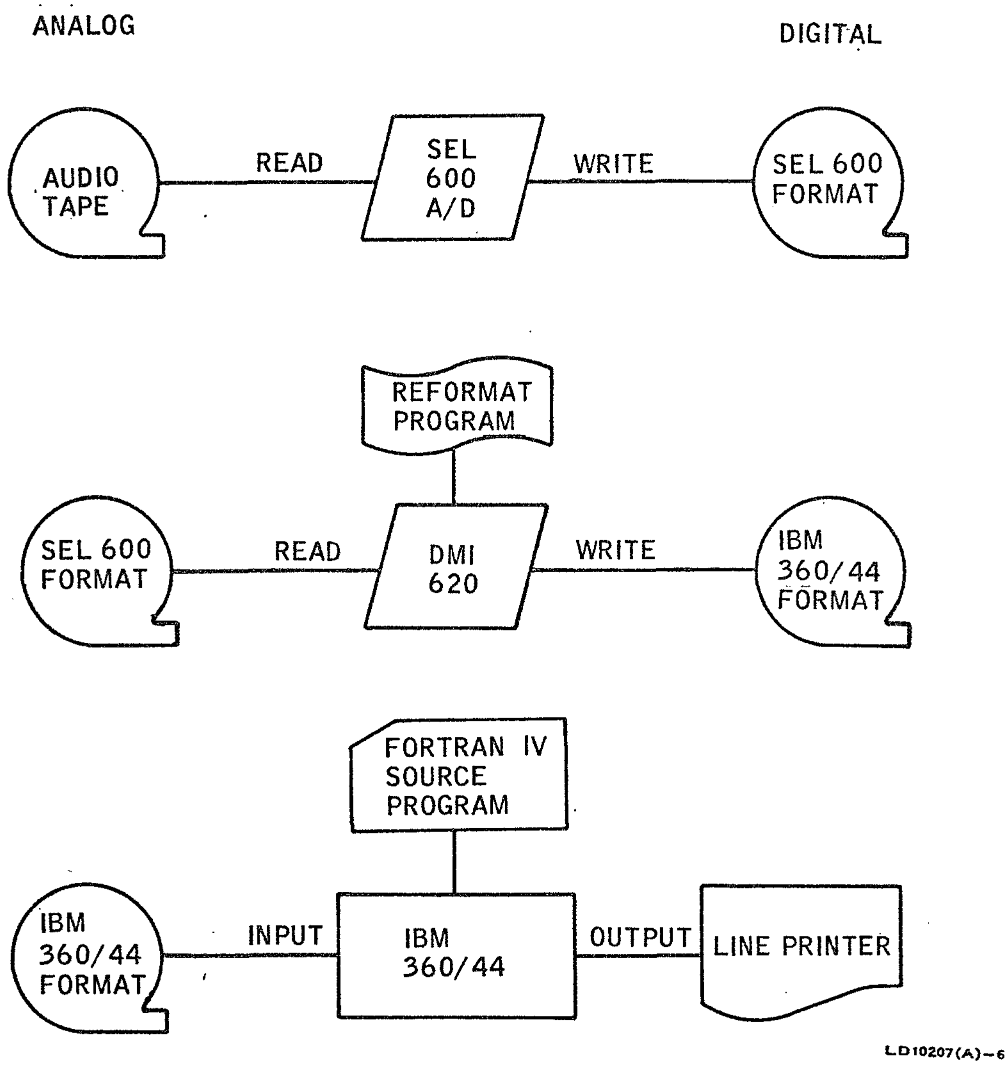

Figure 6 Development Digital Test Configuration for Determining Speech-To-Noise Ratios.

360/44 Software Configuration

The basic development scheme for detection of speech plus noise from noise and the calculation of a SPNR vs. time was written for the IBM 360/44 utilizing FORTRAN IV language. The program consists of four sections.

a. Reading the raw digital data from the tape into the 360 computer.

b. Measuring the raw data and sorting the data into speech plus noise and noise arrays.

c. Performing the SPNR calculations.

d. Writing the outputs with the line printer.

A general explanation of the basic FORTRAN IV program for the IBM S/360/44 is presented in Figure 7, and described below:

Figure 8 is an example of the computer printout.

Position Numbers	
1---23	This area consists of reading information into the computer from the magnetic tape. The information read in consists of 360 sample data points (23.8 msec of data sampled at 15,120 samples/sec.).
24---37	A mean square value of the 23.8 msec data interval is calculated. The number of data intervals are counted and the process continues until 42 mean square values are calculated. This represents 1 second of raw data.
38---54	A counter accumulates the number of seconds of real-time data that has been processed. The 42 values in the mean square array are compared

with their adjacent values for 3 consecutive agreements within 1 dB. An agreement array is established that contains values that are the average of three consecutive agreements. If the number of agreement values does not exceed 1, the program cycles back to the next second of raw data and the computer prints the number registered in the seconds counter and "No Agreements." There must be at least 2 agreements to calculate a SPNR.

55---63 This subroutine sorts the agreement array into algebraic ascending order.

64---76 The first value, or smallest value in the agreement array, is defined as noise and placed in a new array that will contain the noise values. All the values that are within 1 dB of the noise reference are placed into the noise array. All values that are 3 dB or greater than the noise reference are placed in a speech plus noise array.

77---104 If there are no values placed in the speech plus noise array the computer will write the number of seconds in the seconds counter, "Noise Only," and the values in the agreement array. If values are placed in the speech plus noise array, the values of the speech plus noise and noise in the arrays are printed out. The average or mean values of each array are calculated and printed.

105---107 The speech-to-noise ratio is calculated and printed.

108---114 This section of the program tests the number of seconds processed to the required number of seconds desired.

TYPICAL SPNR PROGRAM

```
FORTRAN IV   MODEL 44  PS      VERSION 2    DATE 68        PAGE
                INTEGER*2  IDATA(180)
                  DIMENSION SMQ(45), AGREE(45), SPXN(45), XN(45)
                DIMENSION RASP(180), RASPB(180)
            92  FORMAT (1H1)
            90  FORMAT (90A2,90A2)
                NSEC=0
                NERR=0
                NTOTAL=137
          C       NTOTAL IS THE NUMBER OF SECONDS OF DATA ON THE
                  TAPE
                WRITE (6,92)
          C  TOP OF FORM
                K=0
             2  CONTINUE
             3  READ (2,90,END=4,ERR=300) IDATA
             4  CONTINUE
                I=1
                DO 6 I=1,180
                RASP(I)=IDATA(I)
             6  CONTINUE
                READ (2,90,END=7,ERR=301) IDATA
             7  CONTINUE
                I=1
                DO 8 I=1,180
                RASPB(I)=IDATA(I)
             8  CONTINUE
          C  RASP HAS 180 SAMPLES AND RASPB HAS 180 SAMPLES
                K=K+1
                SMQ(K)=0.0
```

Figure 7 FORTRAN IV Program

```
      I=1
   10 DO 12 I=1,180
   11 SMQ(K)=SMQ(K)+(RASP(I)**2)/360)+(RASPB(I)**2)/360
   12 CONTINUE
C SMQ  IS AVERAGE MEAN SQUARE OF 360 SAMPLES
   13 IF (K-42) 2,14,14
  300 CONTINUE
      NERR=NERR+1
      TO TO 4
  301 CONTINUE
      NERR=NERR+1
      GO TO 7
C GO TO 14 IF WE HAVE 42 AVG MEAN SQUARES
   14 CONTINUE
      NSEC=NSEC+1
   34 CONTINUE
  100    K=0
         DO 110 I=1,40
         COMPRE=SMQ(I)/SMQ(I+1)
         IF(COMPRE-1.259)120,120,110
  120 IF(COMPRE-0.7943)110,130,130
  130 COMPRE=SMQ(I+1)/SMQ(I+2)
      IF(COMPRE-1.259)140,140,110
  140 IF(COMPRE-0.7943)110,150,150
  150 K=K+1
      AGREE(K)=SMQ(I)+SMQ(I+1)SMQ(I+2)/3.0
  110 CONTINUE
C     IF K IS NOT GREATER THAN 1 WE CANNOT SOLVE FOR SPNR
```

Figure 7 (Cont'd) (2 of 5)

```
      IF (K-1)156,156,160
155   FORMAT(10X,I10,'NO AGREEMENTS')
156   WRITE(6,155)NSEC
      GO TO 40
C     SORT THE AGREE ARRAY INTO ALGEBRAICALLY ASCENDING
      ORDER
160   L=K-1
      DO 170 MO=1,L
      NO=K-MO
      DO 170 I=1,NO
      IF(AGREE(I)-AGREE(I+1))170,170,180
180   X=AGREE(I)
      AGREE(I)=AGREE(I+1)
      AGREE(I+1)=X
170   CONTINUE
C     USE FIRST VALUE IN AGREE ARRAY FOR NOISE REFERENCE
      AND SORT
      XN(1)=AGREE(1)
      N=0
      M=1
      DO 190 I=2,K
      SHIFT=AGREE(1)/AGREE(I)
      IF(SHIFT-0.7943)192,210,210
192   IF(SHIFT-0.5012)200,200,190
200   N=N+1
      SPXN(N)=AGREE(I)
      GO TO 190
210   M=M+1
      XN(M)=AGREE(I)
```

Figure 7 (Cont'd) (3 of 5)

```
FORTRAN IV    MODEL  44  PS    VERSION 2   DATE 68             PAGE
        190  CONTINUE
             ASPXN=0
             IF(N-1)236,237,221
        219  FORMAT (1H ,I6,'TH SECOND      SPEECH+NOISE VALUES')
        220  FORMAT (1H ,7F12.1)
        221  WRITE (6,219)NSEC
             WRITE (6,220)(SPXN (I),I=1,N)
        230  FORMAT (1H,'NOISE VALUES     ',7F12.1)
             WRITE(6,230)(XN(I),I=1,M)
      C      FIND AVERAGE VALUE OF SPEECH & NOISE ARRAY AND PRINT
        235  DO 240 I=1,N
             ASPXN=(SPXN(I))/N+(ASPXN)
                GO TO 240
        237     ASPXN=SPXN(1)
        240     CONTINUE
                GO TO 239
        238  FORMAT(1H ,I6,5X,'NOISE ONLY',7F12.1//)
        236     WRITE(6,238)NSEC,(AGREE(I),I=1,K)
             GO TO 40
      C         FIND AVERAGE VALUE OF NOISE ARRAY AND PRINT
        239     AXN=0
                IF(M-1)247,247,245
        245     DO 250 I=1,M
                AXN=(XN(I))/M+(AXN)
                GO TO 250
        247     AXN=XN(1)
        250     CONTINUE
```

Figure 7 (Cont'd) (4 of 5)

```
0101        260      FORMAT(1X,I10,3X,'AVERAGE SPEECH + NOISE IS'
                     ,2X,F12.4//)
0102                 WRITE(6,260)NSEC,ASPXN
0103        270      FORMAT(1X,I10,3X,'AVERAGE NOISE IS',11X,F12.4//)
0104                 WRITE(6,270)NSEC,AXN
          C          CALCULATE THE SPEECH TO NOISE RATIO
0105        280      SPNR=10* ALOG10((ASPXN-AXN)/AXN)
0106        290   FORMAT(1X,I10,3X,'THE SPEECH TO NOISE RATIO IS',
                  2X,F12.4,1X,'DB'/C/)
0107                 WRITE(6,290)NSEC,SPNR
          C  SPNR IS THE SPEECH TO NOISE RATIO
0108         40 K=0
0109            IF (NSEC-NTOTAL) 2,2,60
          C           NSEC IS NUMBER OF SECONDS PROCESSED
0110         60 CONTINUE
0111         97 FORMAT (1H ,I9,' READ ERROR5')
0112            WRITE (6,97)NERR
0113            STOP
0114            END
```

Figure 7 (Cont'd) (5 of 5)

EXAMPLE OF COMPUTER PRINTOUT

	SPEECH + NOISE VALUES	264530.25	280649.93	285905.56	292413.56	295267.06	313165.31	319454.06
		320226.43	326179.37	329327.62				
	NOISE VALUES	532.9155	561.4314	666.3489	669.2791			
1	AVERAGE SPEECH + NOISE IS		302711.7500					
1	AVERAGE NOISE IS		607.4934					
1	THE SPEECH TO NOISE RATIO IS		26.9961 DB					
2	NOISE ONLY	464.6555	489.1960	494.7708	503.2505	508.2654	515.0093	536.7175
		536.7683	538.3513	546.5356	563.3120	565.3777	568.3245	574.0569

FIGURE 8 COMPUTER PRINTOUT

Verification Testing of SPNR Measurement

The primary objectives of the verification tests that were conducted were to disclose obvious hardware or software problems and to obtain adequate experience and background by working with the developed software scheme and hardware configuration. All tests were conducted with the hardware configuration as shown in Figure 6. Minor software modifications were made throughout various phases of testing (e.g., floating point to integer for the raw data read into the 360/44 and modifications to accept different sampling rates). The tests described below are in the order that they were performed.

Test I

Conditions: Three tones (300 Hz, 1 KHz, and 4 KHz) that were recorded at OVU were played into the system, digitized at 15.12 K samples/sec (2 volts p-p) and their mean square values calculated and printed out.

Results: The mean square values at 1 KHz were down approximately 5 dB and the mean square values at 4 KHz were down over 10 dB with reference to the 300 Hz. The poor frequency response was due to low pass filters that are incorporated in the input of the SEL 600.

Test II

Conditions: The same audio test tape as described in Test I was played directly into the digitizer bypassing the filter of the SEL 600. The mean square values were calculated and printed out as in Test I.

Results: The values at 1 KHZ were down approximately 1 dB and the values at 4 KHz were down approximately 3 dB with reference to the 300 Hz. It was determined that the roll off was due to the Magnecord tape recorder and not the digitizing process and calculations. A decision was made to accept this frequency non-linearity and continue testing by bypassing the input filters.

Test III

Conditions: A test tape was made of two phrases, from four actual systems compatibility test data tapes. The measured signal-to-noise ratios of the 1 KHz tone were approximately 30 dB, 20 dB, 10 dB, and 3 dB. The test tape was played into the system, digitized at 15.12 K samples/sec with the input level not exceeding 2 volts p-p on voice peaks. The development computer program was used to measure SPNR's.

Results: The mean values of SPNR's were 24.1 dB, 12.9 dB, 7.7 dB and 0.8 dB. The mean square values appeared low in relation to the quantization level of 2048 for 11 bits. It was determined at this time that the full scale input of the A/D converter was actually 4 volts p-p rather than 2 volts p-p. The output also indicated that the detection scheme was sensitive to level changes that were caused by sudden shifts of the noise levels. The level shifts were caused during the making of the test tape and did not appear to be a serious problem at the time. A decision was made to see if the SPNR's were sensitive to the input level to the SEL 600.

Test IV

Conditions: Same as in Test III except the input voice peaks were adjusted to just below 4 volts p-p.

Results: The mean values of SPNR's were 25.3 dB, 12.9 dB, 7.7 dB, and 0.8 dB. These results (in comparison to Test III) indicated that the SPNR measurement was not a function of the input level to the converter. Changes of the playback gain affect the speech and noise proportionally as long as the level is in the normal range of the analog recorder. This test also indicated a good repeatability of the SPNR measurement.

The difference of 1.2 dB for the first two phrases was considered neglibible since the speech values have an approximate 12 db peak factor as discussed in the section on speech characteristics and there should be more sensitivity at higher SPNR's. Since the digitizing for Test IV was not time synchronous with the the digitizing for Test III, the results cannot be exact since different values fall in one-second time intervals. It was then decided to test the SPNR measurement at lower sampling rates.

Test V

Conditions: Same as in Test IV except the digitizing rate of the SEL 600 was changed to 7.56 K samples/sec.

Results: The mean values of the SPNR's were 26.1 dB, 12.9 dB, 7.5 dB and 0.197 dB. The results indicated that no significant differences occurred at this digitizing rate. Note that 7.56 K samples/sec is just below the Nyquist frequency for aliasing errors assuming an information or noise bandwidth of 4 KHz. The tapes that were used had frequency components down 35 dB at the upper edge of the band.

Test VI

Conditions: Thirty-three seconds of an EVA demonstration audio tape was tested. For this tape, the speaker had no requirement to speak at a constant level as did the speakers making lab test tapes. This tape was indicative of an actual manned mission tape since the material recorded was continuous and contained spacecraft and checkout procedures. The SEL 600 digitizing rate was 7.56 K samples/sec and the input level was set to just less than 4 volts p-p on the voice peaks.

Results: SPNR's varied over the 33 seconds from a minimum of 18.6 dB to a maximum of 26.2 dB. The mean value over 33 seconds was calculated to be 21.7 dB. The standard deviation was calculated to be 2.8 dB over the same time. It was also noted during this test that in some 1 second intervals the output printed was "Noise Only," however, the noise values printed were actually all speech plus noise values. This indicated that the speaker was able to speak continuously at a fast enough rate as not to allow for least 74 msec pauses to be detected as noise during 1 second intervals. The voice channel that was used also had a voice operated relay (VOX) circuit. The detection scheme in the software used the lowest measured values as noise which gives inaccurate SPNR's during activation and deactivation of the VOX circuit since the VOX level is not the noise that competes with the speech.

V. CONCLUSIONS

The following conclusions can be made as a result of the study presented on the development of a speech-to-noise power ratio measurement, utilizing digital techniques.

The conclusions based on the general requirements of the measurement technique as outlined in Section III are as follows:

- The discrimination between speech and noise can be successfully accomplished using the present system software.
- The noise measurements associated with the SPNR calculations can be obtained during pauses in conversational speech if the pauses exceed approximately 0.1 seconds.
- The elimination of errors in the SPNR measurements, caused by sudden shifts in noise levels, can be realized by adding more intelligence to the present developed software.
- The present measurement method is based on a continuous loop technique in that it can measure time varying speech communication links.
- The processing efficiency of the 360/44 program is approximately 3 to 1 for real-time measurement calculations (i.e., it requires 3 seconds to process 1 second of real-time data).
- The method presently designed for the SPNR measurement is compatible with any format of voice inputs.
- Speech-to-noise ratios can be obtained to permit evaluation of voice communication links during launch and staging phases of manned missions by statistically optimizing the time intervals of the measurement.

The conclusions based on the present developed method of obtaining a SPNR measurement are as follows:

- The dynamic range of the present measurement configuration is from 0 dB to 66 dB.

- A digitized sampling rate of 7.56 K samples/sec is adequate to obtain accurate measurements.

- The measurement is not sensitive to input level adjustments of the A/D converter, that is, as long as there are at least 256 quantization levels to define the full range input.

- Another application of the SPNR measurement is to obtain real-time quality measurements of the voice channel during manned Apollo missions (e.g., when two or more sites are acquiring the downlink voice, a quantitative measure of voice quality can be made to determine which site should be switched to the main intercom bus).

- The hardware available in the Data Systems Development Laboratory of ISD is capable of performing the task of development of the speech-to-noise ratio measurement and further voice analysis work using statistical correlation methods.

- The analog verification system that was configured to simulate the digital technique can be further developed for use in the Telecommunication Audio Lab for quick-look analysis and other applications.

VI. RECOMMENDATIONS

It is recommended that an advanced or Phase II development program be initiated to refine the present measurement method to operational status. The refinement task should be conducted with the present system configuration as shown in Figure 6. Required software changes can be expediently made using FORTRAN IV language and utilizing the computer's library of subroutines. When the Scientific Data Systems (SDS) A/D converter is interfaced with the DMI 620 System, it will eliminate a time consuming step of digitzing on the SEL 600 and reformatting on the DMI 620. The major areas of effort for the Phase II development program are as follows:

a. Elimination of the level change errors that occur as a result of the present software sorting schemes.

b. Optimization of the following parameters included in the present method: number of consecutive agreements required, time of comparison intervals, required time of raw data for valid SPNR vs time measurements.

c. Evaluation of the validity of the finalized method taking into account statistical probability analysis.

d. Measuring representative amounts of audio tapes, using the finalized method on a link basis, to determine curves of SPNR vs WI scores.

It is also recommended that another digital system be developed, as a parallel task, for voice analysis studies. The configuration of this system is shown in Figure 9. The requirements of this proposed system will be to complement the present measurement approach and to develop additional digital voice analysis techniques. The primary component in this proposed system is the Astrodata High-Speed Statistical Calculation Unit. This component is designed to function in conjunction with the DMI 620 as a statistical correlation system.

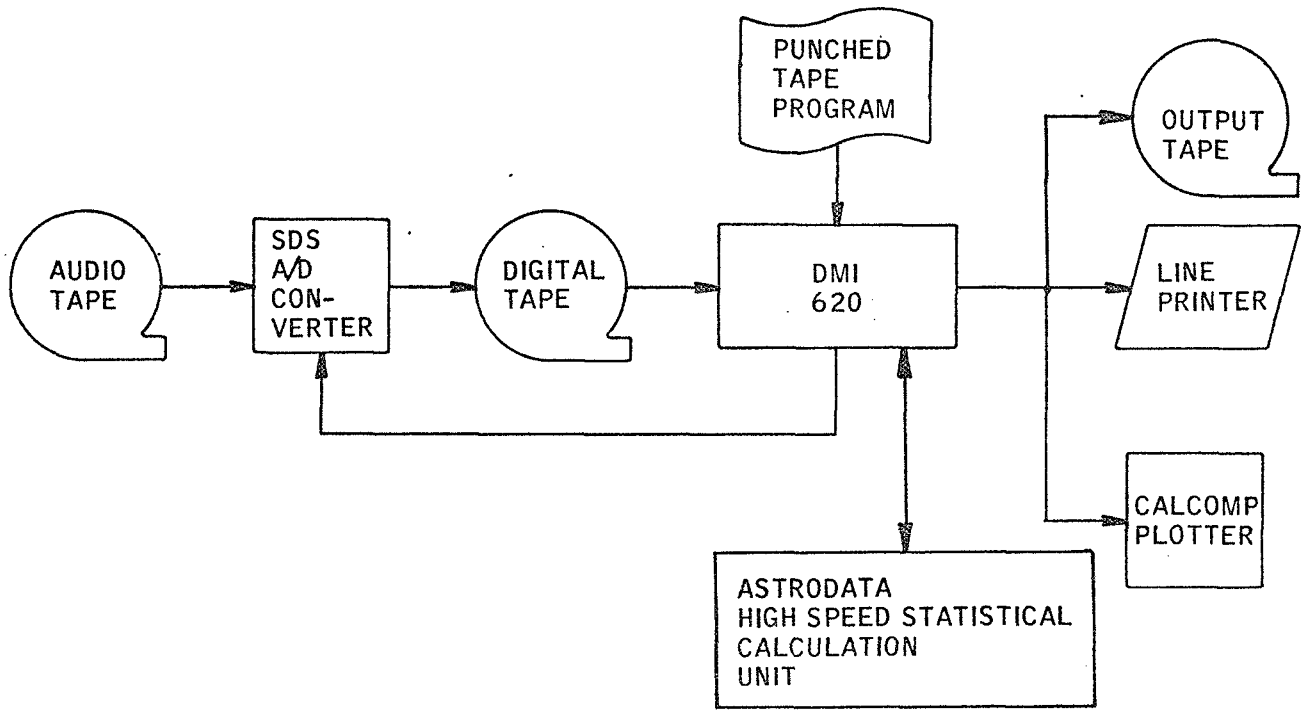

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Figure 9 Statistical Correlation System Configuration

Of main interest is the degradation of Apollo voice communications due to noise masking of the speech spectra, amplitude distortion caused by peak and center clipping and frequency distortion. Weighting factors can be applied to the measured speech-to-noise ratios to take into account these non linearities. Analyses can be made of these effects utilizing the proposed system and the following statistical calculations:

a. Auto-correlation

b. Cross-correlation

c. Power Spectral density

d. Fourier analysis

e. Cross power spectral density

APPENDIX A

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APPENDIX B

THEORY OF RANDOM DATA MEASUREMENTS

MEASUREMENTS AND STATISTICAL ANALYSIS

In any study of the statistics of speech waves, it is assumed that these waves are the result of a stochastic (e.g., chance) process. It is preferrable to confine the analysis of these waves to a time interval where sensible stationary statistics may be obtained. The advantages are the ability to ignore any reference to an absolute time origin and to interchange the processes of statistical and time averaging. It is implied here that the concern is with only the long-time statistics; that is, the statistics of speech in its entirety, rather than with the statistics of individual speech sounds.

Descriptive Properties of Random Data

Mean Square Values:

The general intensity of any random data may be described by a mean square value, which is the average of the squared values of the time history.

$$\Psi_x^2 = \lim_{T\to\infty} \frac{1}{T} \int_0^T x^2(t)dt \qquad (1)$$

The positive square root of the mean square value is called the root mean square or rms value.

It is often desirable to think of physical data in terms of the combination of a static or time-invariant component and a dynamic or fluctuating component. The static component may be described by a mean value that is the average of all values. In equation form the mean value μ_x is given by,

$$\mu_x = \lim_{T\to\infty} \frac{1}{T} \int_0^T x(t)dt \qquad (2)$$

The dynamic component may be described by a variance which is simply the mean square value about the mean. In equation form, the variance σ_x^2 is given by,

$$\sigma_x^2 = \lim_{T\to\infty} \frac{1}{T} \int_0^T [x(t) - \mu_x]^2 dt \tag{3}$$

The positive square root of the variance is the standard deviation. By expanding the last equation it is apparent that the variance is equal to the mean square minus the square of the mean value. That is,

$$\sigma_x^2 = \Psi_x^2 - \mu_x^2 \tag{4}$$

Probability Density Functions

The probability density function for random data describes the probability that the data will assume a value within some defined range of time.

$$P(x) = \lim_{\Delta x\to o} \lim_{T\to\infty} \frac{1}{T} \left(\frac{Tx}{\Delta x}\right) \tag{5}$$

where Tx is the total amount of time that x(t) falls inside the range Δx during an observation time T. The principle application for probability density function measurement of physical data is to establish a probability description for the instantaneous values of the data. However, it can also be used to distinguish between sinusoidal and random data due to the differences in resulting probability density function plots.

Autocorrelation Functions

The autocorrelation function for random data describes the general dependence of the data values at one time interval to the values at a future time interval. An estimate for the autocorrelation between the values of x(t) at times t and t + τ may be obtained by taking the product of the two values and averaging over the observation time T. The resulting average product will approach an exact autocorrelation function as T approaches infinity.

In equation form,

$$R_x(\tau) = \lim_{T\to\infty} \frac{1}{T}\int_0^T x(t)\ x\ (t+\tau)dt \tag{6}$$

In terms of the autocorrelation function, the mean value of x(t) is given (excluding such special cases as sine waves) by,

$$\mu_x = \sqrt{R_x(\infty)} \tag{7}$$

That is, the mean value of x(t) is equal to the positive square root of the autocorrelation as the time displacement becomes very long. Similarly, the mean square value of x(t) is given by,

$$\Psi_x{}^2 = R_x(0) \tag{8}$$

That is, the mean square value is equal to the autocorrelation at zero time displacement.

The principle application for the autocorrelation function measurement of physical data is to establish the influence of values at any time over values at a future time. Because a sine wave, or any other deterministic data, will have an autocorrelation function that persists over all time displacements, as opposed to random data that diminishes to zero for large time displacements (assuming $\mu_x = 0$). An autocorrelation measurement provides a powerful tool for detecting deteriministic data that may be masked in a random background.

<u>Power Spectral Density Functions</u>

The power spectral density functions for random data describes the general frequency composition of the data in terms of the spectral density of the mean square value. The mean square value of a sample time history record in a frequency range between f and f + Δf may be obtained by: (1). filtering the sample record with a bandpass filter having sharp cut-off characteristics and (2). computing the average of the squared output from the filter. The average squared value will approach an exact mean square value as the ovservation time T approaches infinity. The equation form,

$$\Psi_x^2\ [f,\ f+\Delta f] = \lim_{T\to\infty} \frac{1}{T} \int_0^T x^2(t,\ f,\ \Delta f)dt \tag{9}$$

where x(t, f, Δf) is that portion of x(t) in the frequency range from f to f + Δf).

The principal application for a power spectral density function measurement of physical data is to establish the frequency composition of data that indicates important relationships to the basic characteristics of the physical system involved.

DIGITAL COMPUTER TECHNIQUES

Digitizing of Continuous Data

The process of digitizing consists of converting continuous data into discrete numbers. The two main parts involved in a digitization procedure are sampling and quantization. It is important to have a sufficient number of samples to properly describe the significant information in the high frequencies. However, sampling at points that are too close together will yield correlated and highly redundant data and increase greatly both the labor and cost of calculations. To reduce the number of samples, the sampling rate should be reduced to the lowest rate that will avoid aliasing errors. Aliasing constitutes an error that does not occur in analog processing of data. If the time interval T between samples is h seconds, then the sampling rate is 1/h samples per second. The useful data will be from 0 to 1/2 h cycles per seconds since frequencies in the data that are higher than 1/2 h cycles per second will fold into the lower frequency range. The cutoff frequency,

$$f_c = \frac{1}{2h} \tag{10}$$

is known as the Nyquist frequency.

If it is assumed that the quantization error follows a uniform probability over one scale unit, then these errors will have a mean value of zero and a standard deviation of approximately 0.29 scale unit. This is the rms value of the quantization error, which may be considered as an rms noise on desired signals.

$$\text{RMS Signal-To-Noise Ratio} = \frac{\text{Full Range Signal}}{0.29} \tag{11}$$

Calculation of the Mean Square Value

The sample mean square values is given by,

$$\overline{X}^2 = \frac{1}{N}\sum_{n=1}^{N} (X_n)^2 \tag{12}$$

where N is the number of data samples and X_n is the transformed data values with $\overline{X} = 0$.